Voice Over IP for the Cisco 3600 Series Commands

This chapter documents new commands. All other commands used with this feature are documented in the Cisco IOS Release 11.3 command reference documents.

The following new commands are used to configure and monitor Voice over IP:

- **acc-qos**
- **answer-address**
- **codec**
- **comfort-noise**
- **connection**
- **cptone**
- **description**
- **destination-pattern**
- **dial-control-mib**
- **dial-peer voice**
- **dial-type**
- **echo-cancel coverage**
- **echo-cancel enable**
- **expect-factor**
- **fax-rate**
- **icpif**
- **impedance**
- **input gain**
- **ip precedence**
- **ip udp checksum**
- **music-threshold**
- **non-linear**
- **num-exp**
- **operation**

- **output attenuation**
- **port**
- **prefix**
- **req-qos**
- **ring frequency**
- **ring number**
- **session-protocol**
- **session-target**
- **show call active voice**
- **show call history voice**
- **show dial-peer voice**
- **show dialplan incall number**
- **show dialplan number**
- **show num-exp**
- **show voice port**
- **shutdown** (dial-peer configuration)
- **shutdown** (voice-port configuration)
- **signal**
- **snmp enable peer-trap poor-qov**
- **snmp-server enable traps**
- **snmp trap link-status**
- **timeouts initial**
- **timeouts interdigit**
- **timing clear-wait**
- **timing delay-duration**
- **timing delay-start**
- **timing dial-pulse min-delay**
- **timing digit**
- **timing interdigit**
- **timing pulse**
- **timing pulse-interdigit**
- **timing wink-duration**
- **timing wink-wait**
- **type**
- **vad**
- **voice-port**

A subset of the commands listed are voice-port commands. Voice-port commands are supported on different voice signaling types, which vary depending on the platform. Table 4-1 lists the Cisco 3600 series voice port commands and the signaling types supported.

Table 4-1 Cisco 3600 Series Commands and Signaling Types Supported

Voice Port Command	FXO	FXS	E&M
comfort-noise			
connection			
cptone	X	X	X
description	X	X	X
dial-type	X		X
echo-cancel coverage			
echo-cancel enable			
impedance	X	X	X
input gain	X	X	X
music-threshold			
non-linear			
operation			X
output attenuation	X	X	X
ring frequency		X	
ring number	X		
shutdown	X	X	X
signal	X	X	X
snmp trap link-status			
timeouts initial			
timeouts interdigit			
timing			
timing clear-wait			X
timing delay-duration			X
timing delay-start			X
timing dial-pulse min-delay			X
timing digit	X	X	X
timing inter-digit	X	X	X
timing pulse	X		X
timing pulse-inter-digit	X		X
timing wink-duration			X
timing wink-wait			X
type			X

Command Syntax Conventions

Table 4-2 describes the syntax used with the commands in this chapter.

Table 4-2 Command Syntax Guide

<table>
<tr><th>Convention</th><th>Description</th></tr>
<tr><td>boldface font</td><td>Commands and keywords.</td></tr>
<tr><td>italic font</td><td>Command input that is supplied by you.</td></tr>
<tr><td>[ ]</td><td>Keywords or arguments that appear within square brackets are optional.</td></tr>
<tr><td>{ x | x | x }</td><td>A choice of keywords (represented by x) appears in braces separated by vertical bars. You must select one.</td></tr>
<tr><td>^ or Ctrl</td><td>Represent the key labeled Control. For example, when you read ^D or Ctrl-D, you should hold down the Control key while you press the D key.</td></tr>
<tr><td><code>screen font</code></td><td>Examples of information displayed on the screen.</td></tr>
<tr><td><code>boldface screen font</code></td><td>Examples of information that you must enter.</td></tr>
<tr><td>< ></td><td>Nonprinting characters, such as passwords, appear in angled brackets.</td></tr>
<tr><td>[ ]</td><td>Default responses to system prompts appear in square brackets.</td></tr>
<tr><td>Note</td><td>Means reader take note. Notes contain helpful suggestions or references to additional information and material.</td></tr>
<tr><td>Caution</td><td>Means reader be careful. In this situation, you might do amoething that could result in equipemnt damage or loss of data.</td></tr>
<tr><td>Timesaver</td><td>Means the described action saves time. You can save time by performing the action described in the paragraph.</td></tr>
</table>

acc-qos

To generate an SNMP event if the quality of service for a dial peer drops below a specified level, use the **acc-qos** dial-peer configuration command. Use the **no** form of this command to use the default value for this feature.

acc-qos {**best-effort** | **controlled-load** | **guaranteed-delay**}
no acc-qos

Syntax Description

best-effort	Indicates that Resource Reservation Protocol (RSVP) makes no bandwidth reservation.
controlled-load	Indicates that RSVP guarantees a single level of preferential service, presumed to correlate to a delay boundary. The controlled load service uses admission (or capacity) control to assure that preferential service is received even when the bandwidth is overloaded.
guaranteed-delay	Indicates that RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queuing if the bandwidth reserved is not exceeded.

Default

The default value is best-effort. Using the **no** form of this command is the same as the default.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **acc-qos** dial-peer command to generate an SNMP event if the quality of service for specified dial peer drops below the specified level. When a dial peer is used, the Cisco IOS software reserves a certain amount of bandwidth so that the selected quality of service can be provided. Cisco IOS software uses Resource Reservation Protocol (RSVP) to request quality of service guarantees from the network.

To select the most appropriate value for this command, you need to be familiar with the amount of traffic this connection supports and what kind of impact you are willing to have on it. The Cisco IOS software generates a trap message when the bandwidth required to provide the selected quality of service is not available.

This command is only applicable to VoIP peers.

Example

The following example selects guaranteed-delay as the specified level below which an SNMP trap message will be generated:

```
dial-peer voice 10 voip
 acc-qos guaranteed-delay
```

Related Commands

You can use the master index or search online to find documentation of related commands.

req-qos

answer-address

To specify the full E.164 telephone number to be used to identify the dial peer of an incoming call, use the **answer-address** dial-peer configuration command. Use the **no** form of this command to disable this feature.

answer-address [+]*string*
no answer-address

Syntax Description

string	Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are: • Digits 0 through 9, letters A through D, pound sign (#), and asterisk (*), which represent specific digits that can be entered. • Plus sign (+), which is optionally used as the first digit to indicate an E.164 standard number. • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit.

Default

The default value is enabled with a null string.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **answer-address** command to identify the origin (or dial peer) of incoming calls from the IP network. Cisco IOS software identifies the dial peers of a call in one of two ways: either by identifying the interface through which the call is received or through the telephone number configured with the **answer-address** command. In the absence of a configured telephone number, the peer associated with the interface will be associated with the incoming call.

For calls coming in from a POTS interface, the **answer-address** command is not used to select an incoming dial peer. The incoming POTS dial peer is selected on the basis of the port configured for that dial peer.

This command is applicable to both VoIP and POTS dial peers.

Note The Cisco IOS software does not check the validity of the E.164 telephone number; it will accept any series of digits as a valid number.

Example

The following example configures the E.164 telephone number, "555-9626" as the dial peer of an incoming call:

```
dial-peer voice 10 pots
 answer-address +5559626
```

Related Commands

You can use the master index or search online to find documentation of related commands.

destination-pattern
port
prefix

codec

To specify the voice coder rate of speech for a dial peer, use the **codec** dial-peer configuration command. Use the **no** form of this command to reset the default value for this command.

codec {**g711alaw** | **g711ulaw** | **g729r8**}
no codec

Syntax Description

g711alaw	G.711 A-Law 64,000 bits per second (bps).
g711ulaw	G.711 u-Law 64,000 bps.
g729r8	G.729 8000 bps.

Default

The default value for this command is **g729r8**.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **codec** command to define a specific voice coder rate of speech for a dial peer.

For toll quality, use **g711alaw** or **g711ulaw**. These values provide high-quality voice transmission but use a significant amount of bandwidth. For almost toll quality (and a significant savings in bandwidth), use the **g729r8** value.

If **codec** values for the VoIP peers of a connection do not match, the call will fail.

This command is only applicable to VoIP peers.

Example

The following example configures a voice coder rate that provides toll quality and uses a relatively high amount of bandwidth:

```
dial-peer voice 10 voip
 codec g711alaw
```

comfort-noise

To specify whether or not background noise should be generated, use the **comfort-noise** voice-port configuration command. Use the **no** form of this command to disable this feature.

comfort-noise
no comfort-noise

Syntax Description

This command has no arguments or keywords.

Default

The default value for this command is enabled.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **comfort-noise** command to generate background noise to fill silent gaps during calls if VAD is activated. If **comfort noise** is not enabled, and VAD is enabled at the remote end of the connection, the user will hear dead silence when the remote party is not speaking.

The configuration of **comfort noise** only affects the silence generated at the local interface; it does not affect the use of VAD on either end of the connection, or the silence generated at the remote end of the connection.

Example

The following example enables background noise:

```
voice-port 1/0/0
 comfort-noise
```

Related Commands

You can use the master index or search online to find documentation of related commands.

vad

connection

To specify a connection mode for a specified voice port, use the **connection** voice-port configuration command. Use the **no** form of this command to disable the selected connection mode.

connection plar *string*
no connection plar *string*

Syntax Description

plar	Specifies a private line auto ringdown (PLAR) connection. PLAR is handled by associating a peer directly with an interface; when an interface goes off-hook, the peer is used to set up the second call leg and conference them together without the caller having to dial any digits.
string	Specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.

Default

The default value for this command is no connection.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **connection** command to specify a connection mode for a specific interface. Use the **connection plar** command to specify a PLAR interface. The string you configure for this command is used as the called number for all calls coming in over this voice port. The destination dial peer is determined on the basis of this called number.

If the **connection** command is not configured, the standard session application outputs a dial tone when the interface goes off-hook until enough digits are collected to match a dial-peer and complete the call.

Example

The following example selects PLAR as the connection mode, with a destination telephone number of 555-9262:

```
voice-port 1/0/0
 connection plar 5559262
```

Related Commands

You can use the master index or search online to find documentation of related commands.

session-protocol

cptone

To configure a voice call progress tone locale, use the **cptone** voice-port configuration command. Use the **no** form of this command to disable this feature.

cptone {**australia** | **brazil** | **china** | **france** | **germany** | **japan** | **northamerica** | **unitedkingdom**}
no cptone

Syntax Description

australia	Specifies an analog voice interface-related default tone, ring, and cadence setting for Australia.
brazil	Specifies an analog voice interface-related default tone, ring, and cadence setting for Brazil.
china	Specifies an analog voice interface-related default tone, ring, and cadence setting for China.
finland	Specifies an analog voice interface-related default tone, ring, and cadence setting for Finland.
france	Specifies an analog voice interface-related default tone, ring, and cadence setting for France.
germany	Specifies an analog voice interface-related default tone, ring, and cadence setting for Germany.
japan	Specifies an analog voice interface-related default tone, ring, and cadence setting for Japan.
northamerica	Specifies an analog voice interface-related default tone, ring, and cadence setting for North America.
unitedkingdom	Specifies an analog voice interface-related default tone, ring, and cadence setting for the United Kingdom.

Default

The default value for this command is **northamerica**.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **cptone** command to specify a regional analog voice interface-related tone, ring, and cadence setting for a specified voice port. This command only affects the tones generated at the local interface. It does not affect any information passed to the remote end of a connection, or any tones generated at the remote end of a connection.

Example

The following example configures North America as the call progress tone locale:

```
voice-port 1/0/0
 cptone northamerica
```

description

To include a description of what this voice port is connected to, use the **description** voice-port configuration command. Use the **no** form of this command to disable this feature.

description *string*
no description

Syntax Description

string	Character string from 1 to 255 characters.

Default

The default for this command is enabled with a null string.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **description** command to include descriptive text about this voice-port connection. This information is displayed when you enter a **show** command and does not affect the operation of the interface in any way.

Example

The following example identifies this voice port as being connected to the Purchasing department:

```
voice-port 1/0/0
 description purchasing_dept
```

destination-pattern

To specify either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer, use the **destination-pattern** dial-peer configuration command. Use the **no** form of this command to disable this feature.

destination-pattern [+]*string*
no destination-pattern

Syntax Description

string	Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are: • Digits 0 through 9, letters A through D, pound sign (#), and asterisk (*), which represent specific digits that can be entered. • Plus sign (+), which is optionally used as the first digit to indicate an E.164 standard number. • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit.

Default

The default value for this command is enabled with a null string.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **destination-pattern** command to define the E.164 telephone number for this dial peer. This pattern is used to match dialed digits to a dial peer. The dial peer is then used to complete the call.

This command is applicable to both VoIP and POTS dial peers.

Note The Cisco IOS software does not check the validity of the E.164 telephone number; it will accept any series of digits as a valid number.

Example

The following example configures the E.164 telephone number, “555-7922,” for a dial peer:

```
dial-peer voice 10 pots
 destination-pattern +5557922
```

Related Commands

You can use the master index or search online to find documentation of related commands.

answer-address
prefix

dial-control-mib

To specify attributes for the call history table, use the **dial-control-mib** global configuration command.

dial-control-mib {**max-size** *number* | **retain-timer** *number*}

Syntax Description

max-size *number*	Specifies the maximum size of the call history table. Valid entries are from **0** to **500** table entries. A value of 0 will prevent any history from being retained.
retain-timer *number*	Specifies the length of time, in minutes, for entries in the call history table. Valid entries are from **0** to **2147483647** minutes. A value of 0 will prevent any history from being retained.

Default

The default call history table length is 50 table entries. The default retain timer is 15 minutes.

Command Mode

Global configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Example

The following example configures the call history table to hold 400 entries, with each entry remaining in the table for 10 minutes:

```
configure terminal
 dial-control-mib max-size 400
 dial-control-mib retain-timer 10
```

dial-peer voice

To enter the dial-peer configuration mode (and specify the method of voice-related encapsulation), use the **dial-peer voice** global configuration command.

dial-peer voice *number* {**voip** | **pots**}

Syntax Description

number	Digit(s) defining a particular dial peer. Valid entries are from 1 to 2147483647.
voip	Indicates that this is a VoIP peer using voice encapsulation on the POTS network.
pots	Indicates that this is a POTS peer using Voice over IP encapsulation on the IP backbone.

Command Mode

Global configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **dial-peer voice** global configuration command to switch to the dial-peer configuration mode from the global configuration mode. Use the **exit** command to exit the dial-peer configuration mode and return to the global configuration mode.

Example

The following example accesses the dial-peer configuration mode and configures a POTS peer identified as dial peer 10:

```
configure terminal
 dial-peer voice 10 pots
```

Related Commands

You can use the master index or search online to find documentation of related commands.

voice-port

dial-type

To specify the type of out-dialing for voice port interfaces, use the **dial-type** voice-port configuration command. Use the **no** form of this command to disable this feature.

dial-type {**dtmf** | **pulse**}
no dial-type

Syntax Description

dtmf	Specifies a touch-tone dialer.
pulse	Specifies a pulse dialer.

Default

The default value for this command is **dtmf**.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **dial-type** command to specify an out-dialing type for an FXO or E&M voice port interface; this command is not applicable to FXS voice ports because they do not generate out-dialing. Voice ports can always detect dtmf and pulse signals. This command does not affect voice port dialing detection.

The **dial-type** command affects out-dialing as configured for the dial peer.

Example

The following example configures a voice port to support a touch-tone dialer:

```
voice-port 1/0/0
 dial-type dtmf
```

echo-cancel coverage

To adjust the size of the echo cancel, use the **echo-cancel coverage** voice-port configuration command. Use the **no** form of this command to reset this command to the default value.

echo-cancel coverage *value*
no echo-cancel coverage *value*

Syntax Description

value	Number of milliseconds the echo-canceller will cover on a given signal. Valid values are 16, 24, and 32.

Default

The default value for this command is 16.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **echo-cancel coverage** command to adjust the coverage size of the echo canceller. This command enables cancellation of voice that is sent out the interface and received back on the same interface within the configured amount of time. If the local loop (the distance from the analog interface to the connected equipment producing the echo) is longer, the configured value of this command should be extended.

If you configure a longer value for this command, it will take the echo canceller longer to converge; in this case, the user might hear slight echo when the connection is initially set up. If the configured value for this command is too short, the user might hear some echo for the duration of the call because the echo canceller is not cancelling the longer delay echoes.

There is no echo or echo cancellation on the IP side of the connection.

Note This command is valid only if the echo cancel feature has been enabled. For more information, refer to the **echo-cancel enable** command.

Example

The following example adjusts the size of the echo canceller to 16 milliseconds:

```
voice-port 1/0/0
 echo-cancel enable
 echo-cancel coverage 16
```

Related Commands

You can use the master index or search online to find documentation of related commands.

echo-cancel enable

echo-cancel enable

To enable the echo cancel feature, use the **echo-cancel enable** voice-port configuration command. Use the **no** form of this command to disable this feature.

echo-cancel enable
no echo-cancel enable

Syntax Description

This command has no arguments or keywords.

Default

The default value for this command is enabled for all interface types.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The **echo-cancel** command enables cancellation of voice that is sent out the interface and is received back on the same interface. Disabling echo cancellation might cause the remote side of a connection to hear an echo. Because echo cancellation is an invasive process that can minimally degrade voice quality, this command should be disabled if it is not needed.

The **echo-cancel** command does not affect the echo heard by the user on the analog side of the connection.

There is no echo path for a 4-wire E&M interface. The echo canceller should be disabled for that interface type.

Note This command is valid only if the **echo-cancel coverage** command has been configured. For more information, refer to the **echo-cancel coverage** command.

Example

The following example enables the echo cancel feature for 16-millisecond echo coverage:

```
voice-port 1/0/0
 echo-cancel enable
 echo-cancel coverage 16
```

Related Commands

You can use the master index or search online to find documentation of related commands.

echo-cancel coverage
non-linear

expect-factor

To specify when the router will generate an alarm to the network manager, indicating that the expected quality of voice has dropped, use the **expect-factor** dial-peer configuration command. Use the **no** form of this command to reset the default value for this command.

expect-factor *value*
no expect-factor *value*

Syntax Description

value	Integers that represent the ITU specification for quality of voice as described in G.113. Valid entries are from 0 to 20, with 0 representing toll quality.

Default

The default value for this command is 10.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Voice over IP monitors the quality of voice received over the network. Use the **expect-factor** command to specify when the router will generate an SNMP trap to the network manager.

This command is only applicable to VoIP peers.

Example

The following example configures toll quality of voice when connecting to a dial peer:

```
dial-peer voice 10 voip
 expect-factor 0
```

fax-rate

To establish the rate at which a fax will be sent to the specified dial peer, use the **fax-rate** dial-peer configuration command. Use the **no** form of this command to reset the default value for this command.

fax-rate{**2400** | **4800** | **7200** | **9600** | **14400** | **disable** | **voice**}
no fax-rate

Syntax Description

2400	Specifies a fax transmission speed of 2400 bits per second (bps).
4800	Specifies a fax transmission speed of 4800 bps.
7200	Specifies a fax transmission speed of 7200 bps.
9600	Specifies a fax transmission speed of 9600 bps.
14400	Specifies a fax transmission speed of 14,400 bps.
disable	Disables fax relay transmission capability.
voice	Specifies the highest possible transmission speed allowed by voice rate.

Default

The default value for this command is **voice**.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **fax-rate** command to specify the fax transmission rate to the specified dial peer.

The values for this command apply only to the fax transmission speed and do not affect the quality of the fax itself. The higher values provide a faster transmission speed but monopolize a significantly larger portion of the available bandwidth. Slower transmission speeds use less bandwidth.

If the **fax-rate** command is set above the CODEC value in the same dial peer, the data sent over the network for fax transmission will be above the bandwidth reserved for RVSP. Because more network bandwidth will be monopolized by the fax transmission, we do not recommend setting the **fax-rate** value higher than the **codec** value. If the **fax-rate** value is set lower than the **codec** value, faxes will take longer to transmit but will use less bandwidth.

This command is only applicable to VoIP peers.

Example

The following example configures a facsimile rate of 9600 bps for faxes sent to a dial peer:

```
dial-peer voice 10 voip
 fax-rate 9600
```

Related Commands

You can use the master index or search online to find documentation of related commands.

codec

icpif

To specify the Calculated Planning Impairment Factor (ICPIF) for calls sent by a dial peer, use the **icpif** dial-peer configuration command. Use the **no** form of this command to restore the default value for this command.

icpif *number*
no icpif *number*

Syntax Description

number	Integer, expressed in equipment impairment factor units, specifying the ICPIF value. Valid entries are 0 to 55.

Default

The default value for this command is 30.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **icpif** command to specify the maximum acceptable impairment factor for the voice calls sent by the selected dial peer.

This command is applicable only to VoIP peers.

Example

The following example disables the **icpif** command:

```
dial-peer voice 10 voip
 icpif 0
```

impedance

To specify the terminating impedance of a voice port interface, use the **impedance** voice-port configuration command. Use the **no** form of this command to restore the default value.

impedance {**600c** | **600r** | **900c** | **complex1** | **complex2**}
no impedance

Syntax Description

600c	Specifies 600 Ohms complex.
600r	Specifies 600 Ohms real.
900c	Specifies 900 Ohms complex.
complex1	Specifies Complex 1.
complex2	Specifies Complex 2.

Default

The default value for this command is 600.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **impedance** command to specify the terminating impedance of an FXO voice-port interface. The impedance value selected needs to match the specifications from the specific telephony system to which it is connected. Different countries often have different standards for impedance. CO switches in the United States are predominantly 600r. PBXes in the United States are normally either 600r or 900c.

If the impedance is set incorrectly (if there is an impedance mismatch), there will be a significant amount of echo generated (which could be masked if the **echo-cancel** command has been enabled). In addition, gains might not work correctly if there is an impedance mismatch.

Configuring the impedance on a voice port will change the impedance on both voice ports of a VNM card. This voice port must be shut down and then opened for the new value to take effect.

This command is applicable to FXS, FXO, and E&M voice ports.

Example

The following example configures an FXO voice port for a terminating impedance of 600 Ohms:

```
impedance 600r
```

input gain

To configure a specific input gain value, use the **input gain** voice-port configuration command. Use the **no** form of this command to disable this feature.

input gain *value*
no input gain *value*

Syntax Description

value	Specifies, in decibels, the amount of gain to be inserted at the receiver side of the interface. Acceptable value is any integer from -6 to 14.

Default

The default value for FXO, FXS, and E&M ports is 0.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

A system-wide loss plan must be implemented using both **input gain** and **output attenuation** commands. Other equipment (including PBXes) in the system must be taken into account when creating a loss plan. This default value for this command assumes that a standard transmission loss plan is in effect, meaning that normally, there must be -6 dB attenuation between phones. Connections are implemented to provide -6 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0.

Please note that you can’t increase the gain of a signal going out into the PSTN, but you can decrease it. Therefore, if the voice level is too high, you can decrease the volume by either decreasing the input gain value or by increasing the output attenuation.

You can increase the gain of a signal coming in to the router. If the voice level is too low, you can increase the input gain.

Example

The following example configures a 3-decibel gain to be inserted at the receiver side of the interface:

```
input gain 3
```

Related Commands

You can use the master index or search online to find documentation of related commands.

output attenuation

ip precedence

To set IP precedence (priority) for packets sent by the dial peer, use the **ip precedence** dial-peer configuration command. Use the **no** form of this command to restore the default value for this command.

ip precedence *number*
no ip precedence

Syntax Description

number	Integer specifying the IP precedence value. Valid entries are 0 to 7. A value of 0 means that no precedence (priority) has been set.

Default

The default value for this command is 0.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **ip precedence** command to configure the value set in the IP precedence field when voice data packets are sent over the IP network. This command should be used if the IP link utilization is high and the QoS for voice packets need to have a higher priority than other IP packets. The **ip precedence** command should also be used if RSVP is not enabled, and the user would like to give voice packets a higher priority over other IP data traffic.

This command is applicable to VoIP peers.

Example

The following example sets the IP precedence at 5:

```
dial-peer voice 10 voip
 ip precedence 5
```

ip udp checksum

To calculate the UDP checksum for voice packets transmitted by the dial peer, use the **ip udp checksum** dial-peer configuration command. Use the **no** form of this command to disable this feature.

ip udp checksum
no ip udp checksum

Syntax Description

This command has no arguments or keywords.

Default

The default value for this command is disabled.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **ip udp checksum** command to enable UDP checksum calculation for each of the outbound voice packets. This command is disabled by default to speed up the transmission of the voice packets. If you suspect that the connection has a high error rate, you should enable **ip udp checksum** to prevent bad voice packets forwarded to the DSP.

This command is applicable to VoIP peers.

Example

The following example calculates the UDP checksum for voice packets transmitted by this dial peer:

```
dial-peer voice 10 voip
 ip udp checksum
```

music-threshold

To specify the threshold for on-hold music for a specified voice port, use the **music-threshold** voice-port configuration command. Use the **no** form of this command to disable this feature.

music-threshold *number*
no music-threshold *number*

Syntax Description

number	Specifies the on-hold music threshold in decibels. Valid entries are any integer from -70 to -30.

Default

The default value for this command is -38.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **music-threshold** command to specify the decibel level of music played when calls are put on hold. This command tells the firmware to pass steady data above the specified level. It only affects the operation of VAD when receiving voice.

If the value for this command is set too high, VAD will interpret music-on-hold as silence, and the remote end will not hear the music. If the value for this command is set too low, VAD will compress and pass silence when the background is noisy, creating unnecessary voice traffic.

Example

The following sets the decibel threshold to -35 for the music played when calls are put on hold:

```
voice port 1/0/0
 music-threshold -35
```

non-linear

To enable non-linear processing in the echo canceller, use the **non-linear** voice-port configuration command. Use the **no** form of this command to disable this feature.

non-linear
no non-linear

Syntax Description

This command has no arguments or keywords.

Default

The default for this command is enabled for all voice-port types.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command is associated with the echo canceller operation. The **echo-cancel enable** command must be enabled for the **non-linear** command to take effect. Use the **non-linear** command to shut off any signal if no near-end speech is detected.

Enabling the **non-linear** command normally improves performance, although some users might perceive truncation of consonants at the end of sentences when this command is enabled.

This feature is also generally known as residual echo suppression.

Example

The following example enables non-linear call processing:

```
voice-port 1/0/0
 non-linear
```

Related Commands

You can use the master index or search online to find documentation of related commands.

echo-cancel enable

num-exp

To define how to expand an extension number into a particular destination pattern, use the **num-exp** global configuration command.

num-exp *extension-number expanded-number*

Syntax Description

extension-number	Digit(s) defining an extension number for a particular dial peer.
expanded-number	Digit(s) defining the expanded telephone number or destination pattern for the extension number listed.

Command Mode

Global configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **num-exp** global configuration command to define how to expand a particular set of numbers (for example, an extension number) into a particular destination pattern. With this command, you can map specific extensions and expanded numbers together by explicitly defining each number, or you can define extensions and expanded numbers using variables. You can also use this command to convert 7-digit numbers to numbers containing less than 7 digits.

Use a period (.) as a variable or wild card, representing a single number. Use a separate period for each number you want to represent with a wildcard—meaning that if you want to replace 4 numbers in an extension with wildcards, type in 4 periods.

Examples

The following example expands the extension number 55541 to be expanded to 14085555541:

```
num-exp 65541 14085555541
```

The following example shows how to expand all 5-digit extensions beginning with 5 to append the following numbers at the beginning of the extension number 1408555:

```
num-exp 5.... 1408555....
```

operation

To select a specific cabling scheme for E&M ports, use the **operation** voice-port configuration command. Use the **no** form of this command as an alternative method of configuring 2-wire operation.

operation {**2-wire** | **4-wire**}
no operation {**2-wire** | **4-wire**}

Syntax Description

2-wire	Specifies a 2-wire E&M cabling scheme.
4-wire	Specifies a 4-wire E&M cabling scheme.

Default

2-wire operation

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The **operation** command will only affect voice traffic. Signaling is independent of 2-wire versus 4-wire settings. If the wrong cable scheme is specified, the user might get voice traffic in only one direction.

Configuring the **operation** command on a voice port changes the operation of both voice ports on a VPM card. The voice port must be shut down and then opened again for the new value to take effect.

This command is not applicable to FXS or FXO interfaces because those are, by definition, two-wire interfaces.

Example

The following example specifies that an E&M port uses a 4-wire cabling scheme:

```
voice-port 1/0/0
 operation 4-wire
```

output attenuation

To configure a specific output attenuation value, use the **output attenuation** voice-port configuration command. Use the **no** form of this command to disable this feature.

output attenuation *value*
no output attenuation

Syntax Description

value	Specifies, in decibels, the amount of attenuation at the transmit side of the interface. Acceptable value is any integer from 0 to 14. The default value for FXO, FXS, and E&M ports is 0.

Default

The default value for FXO, FXS, and E&M ports is 0.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

A system-wide loss plan must be implemented using both **input gain** and **output attenuation** commands. Other equipment (including PBXes) in the system must be taken into account when creating a loss plan. This default value for this command assumes that a standard transmission loss plan is in effect, meaning that normally, there must be -6 dB attenuation between phones. Connections are implemented to provide -6 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0.

Please note that you can’t increase the gain of a signal going out into the PSTN, but you can decrease it. Therefore, if the voice level is too high, you can decrease the volume by either decreasing the input gain value or by increasing the output attenuation.

Example

The following example configures a 3-decibel gain to be inserted at the transmit side of the interface:

```
voice-port 1/0/0
 output attenuation 3
```

Related Commands

You can use the master index or search online to find documentation of related commands.

input gain

port

To associate a dial peer with a specific voice-port, use the **port** dial-peer configuration command. Use the **no** form of this command to cancel this association.

port *slot-number*/*subunit-number*/*port*
no port

Syntax Description

slot-number/	Specifies the slot number in the Cisco router where the voice interface card is installed. Valid entries are from 0 to 3, depending on the slot where it has been installed.
subunit-number/	Specifies the subunit on the voice interface card where the voice port is located. Valid entries are 0 or 1.
port	Specifies the voice port. Valid entries are 0 or 1.

Default

No port is configured.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **port** configuration command to associate the designated voice port with the selected dial peer.

This command is used for calls incoming from a telephony interface to select an incoming dial peer and for calls coming from the VoIP network to match a port with the selected outgoing dial peer.

This command is applicable only to POTS peers.

Example

The following example associates a dial peer with voice port 1, which is located on subunit 0, and accessed through port 0:

```
dial-peer voice 10 pots
 port 1/0/0
```

prefix

To specify the prefix of the dialed digits for this dial peer, use the **prefix** dial-peer configuration command. Use the **no** form of this command to disable this feature.

prefix *string*
no prefix

Syntax Description

string	Integers representing the prefix of the telephone number associated with the specified dial peer. Valid numbers are **0** through **9**, and a comma (**,**). Use a comma to include a pause in the prefix.

Default

The default for this command is a null string.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **prefix** command to specify a prefix for a specific dial peer. When an outgoing call is initiated to this dial peer, the **prefix** *string* value is sent to the telephony interface first, before the telephone number associated with the dial peer.

If you want to configure different prefixes for dialed numbers on the same interface, you need to configure different dial peers.

This command is applicable only to POTS peers.

Example

The following example specifies a prefix of “9” and then a pause:

```
dial-peer voice 10 pots
 prefix 9,
```

Related Commands

You can use the master index or search online to find documentation of related commands.

answer-address
destination-pattern

req-qos

To specify the desired quality of service to be used in reaching a specified dial peer, use the **req-qos** dial-peer configuration command. Use the **no** form of this command to restore the default value for this command.

req-qos {**best-effort** | **controlled-load** | **guaranteed-delay**}
no req-qos

Syntax Description

best-effort	Indicates that Resource Reservation Protocol (RSVP) makes no bandwidth reservation.
controlled-load	Indicates that RSVP guarantees a single level of preferential service, presumed to correlate to a delay boundary. The controlled load service uses admission (or capacity) control to assure that preferential service is received even when the bandwidth is overloaded.
guaranteed-delay	Indicates that RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queuing if the bandwidth reserved is not exceeded.

Default

The default value for this command is best-effort. The **no** form of this command restores the default value.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **req-qos** command to request a specific quality of service to be used in reaching a dial peer. Like **acc-qos**, when you enter this command, the Cisco IOS software reserves a certain amount of bandwidth so that the selected quality of service can be provided. Cisco IOS software uses Resource Reservation Protocol (RSVP) to request quality of service guarantees from the network.

This command is applicable only to VoIP peers.

Example

The following example configures guaranteed-delay as the desired (requested) quality of service to a dial peer:

```
dial-peer voice 10 voip
 req-qos guaranteed-delay
```

Related Commands

You can use the master index or search online to find documentation of related commands.

acc-qos

ring frequency

To specify the ring frequency for a specified FXS voice port, use the **ring frequency** voice-port configuration command. Use the **no** form of this command to reset the default value for this command.

ring frequency *number*
no ring frequency

Syntax Description

number	Specifies the ring frequency (Hertz) used in the FXS interface. Valid entries are 25 and 50.

Default

The default value for this command is 25 Hertz.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **ring frequency** command to select a specific ring frequency for an FXS voice port. Use the **no** form of this command to reset the default value for this command, which is **25** Hertz. The ring frequency you select must match the connected equipment. If set incorrectly, the attached phone might not ring or might buzz. In addition, the ring frequency is usually country-dependent and you should take into account the appropriate ring frequency for your area before configuring this command.

This command does not affect ringback, which is the ringing a user hears when placing a remote call.

Example

The following example configures the ring frequency for 50 Hertz:

```
voice-port 1/0/0
 ring frequency 50
```

Related Commands

You can use the master index or search online to find documentation of related commands.

ring number

ring number

To specify the number of rings for a specified FXO voice port, use the **ring number** voice-port configuration command. Use the **no** form of this command to reset the default value for this command.

ring number *number*
no ring number *number*

Syntax Description

number	Specifies the number of rings detected before answering the call. Valid entries are numbers from 1 to 10.

Default

The default value is 1 ring.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **ring number** command to set the maximum number of rings to be detected before answering a call over an FXO voice port. Use the **no** form of this command to reset the default value, which is **1** ring.

Normally, this command should be set to the default so that incoming calls are answered quickly. If you have other equipment available on the line to answer incoming calls, you might want to set the value higher to give the equipment sufficient time to respond. In that case, the FXO interface would answer if the equipment online did not answer the incoming call in the configured number of rings.

This command is not applicable to FXS or E&M interfaces because they do not receive ringing to receive a call.

Example

The following example sets 5 rings as the maximum number of rings to be detected before closing a connection over this voice port:

```
voice port 1/0/0
 ring number 5
```

Related Commands

You can use the master index or search online to find documentation of related commands.

ring frequency

session protocol

To establish a session protocol for calls between the local and remote routers via the packet network, use the **session protocol** dial-peer configuration command. Use the **no** form of this command to reset the default value for this command.

session protocol cisco
no session protocol

Syntax Description

cisco	Specifies Cisco Session Protocol.

Default

The default value for this command is **cisco**.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

For this release, Cisco Session Protocol (**cisco**) is the only applicable session protocol. This command is applicable only to VoIP peers.

Example

The following example selects Cisco Session Protocol as the session protocol:

```
dial-peer voice 10 voip
 session protocol cisco
```

Related Commands

You can use the master index or search online to find documentation of related commands.

session target

session target

To specify a network-specific address for a specified dial peer, use the **session target** dial-peer configuration command. Use the **no** form of this command to disable this feature.

session target {**ipv4:***destination-address* | **dns:**[**s. | d. | u.**] *host-name* | **loopback:rtp** | **loopback:compressed** | **loopback:uncompressed**}
no session target

Syntax Description

ipv4:*destination-address*	IP address of the dial peer.
dns:*host-name*	Indicates that the domain name server will be used to resolve the name of the IP address. Valid entries for this parameter are characters representing the name of the host device. (Optional) You can use 1 of the following 3 wildcards with this keyword when defining the session target for VoIP peers: • **s.**—Indicates that the source destination pattern will be used as part of the domain name. • **d.**—Indicates that the destination number will be used as part of the domain name. • **u.**—Indicates that the unmatched portion of the destination pattern (such as a defined extension number) will be used as part of the domain name.
loopback:rtp	Indicates that all voice data will be looped-back to the originating source. This is applicable for VoIP peers.
loopback:compressed	Indicates that all voice data will be looped-back in compressed mode to the originating source. This is applicable for POTS peers.
loopback:uncompressed	Indicates that all voice data will be looped-back in uncompressed mode to the originating source. This is applicable for POTS peers.

Default

The default for this command is enabled with no IP address or domain name defined.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **session target** command to specify a network-specific address or domain name for a dial peer. Whether you select a network-specific address or a domain name depends on the session protocol you select.

The **session target loopback** command is used for testing the voice transmission path of a call. The loopback point will depend on the call origination and the loopback type selected.

The **session target dns** command can be used with or without the specified wildcards. Using the optional wildcards can reduce the number of VoIP dial-peer session targets you need to configure if you have groups of numbers associated with a particular router.

Example

The following example configures a session target using dns for a host, "voice_router," in the domain "cisco.com":

```
dial-peer voice 10 voip
 session target dns:voice_router.cisco.com
```

The following example configures a session target using dns, with the optional u. wildcard. In this example, the destination pattern has been configured to allow for any four-digit extension, beginning with the numbers 1310555. The optional wildcard **u.** indicates that the router will use the unmatched portion of the dialed number—in this case, the four-digit extension, to identify the dial peer. As in the previous example, the domain is "cisco.com."

```
dial-peer voice 10 voip
 destination-pattern 1310555....
 session target dns:$u$.cisco.com
```

The following example configures a session target using dns, with the optional **d.** wildcard. In this example, the destination pattern has been configured for 13105551111. The optional wildcard d. indicates that the router will use the destination pattern to identify the dial peer in the "cisco.com" domain.

```
dial-peer voice 10 voip
 destination-pattern 13105551111
 session target dns:$d$.cisco.com
```

Related Commands

You can use the master index or search online to find documentation of related commands.

destination-pattern
session protocol

show call active voice

To show the active call table, use the **show call active voice** privileged EXEC command.

show call active voice

Syntax Description

This command contains no arguments or keywords.

Command Mode

Privileged EXEC

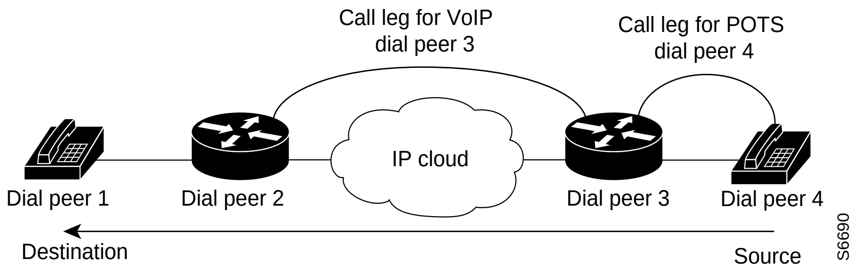
Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **show call active voice** privileged EXEC command to display the contents of the active call table, which shows all of the calls currently connected through the router.

For each call, there are 2 call legs, usually a POTS call leg and a VoIP call leg. A call leg is a discrete segment of a call connection that lies between 2 points in the connection. Each dial peer creates a call leg, as shown in Figure 4-1.

Figure 4-1 Call Legs Example



These 2 call legs are associated by the connection ID. The connection ID is global across the voice network, so that you can associate 2 call legs on one router with 2 call legs on another router, thereby providing an end-to-end view of a call.

Sample Display

The following is sample output from the **show call active voice** command:

```
sloth# show call active voice
GENERIC: SetupTime=21072 Index=0 PeerAddress= PeerSubAddress= PeerId=0
PeerIfIndex=0 LogicalIfIndex=0 ConnectTime=0 CallState=3 CallOrigin=2 ChargedUnits=0
InfoType=0 TransmitPackets=375413 TransmitBytes=7508260 ReceivePackets=377734
ReceiveBytes=7554680

VOIP: ConnectionId[0x19BDF910 0xAF500007 0x0 0x58ED0] RemoteIPAddress=17635075
RemoteUDPPort=16394 RoundTripDelay=0 SelectedQoS=0 SessionProtocol=1
SessionTarget= OnTimeRvPlayout=0 GapFillWithSilence=0 GapFillWithPrediction=600
GapFillWithInterpolation=0 GapFillWithRedundancy=0 HiWaterPlayoutDelay=110
LoWaterPlayoutDelay=64 ReceiveDelay=94 VADEnable=0 CoderTypeRate=0

GENERIC: SetupTime=21072 Index=1 PeerAddress=+14085271001 PeerSubAddress=
PeerId=0 PeerIfIndex=0 LogicalIfIndex=5 ConnectTime=21115 CallState=4 CallOrigin=1
ChargedUnits=0 InfoType=1 TransmitPackets=377915 TransmitBytes=7558300
ReceivePackets=375594 ReceiveBytes=7511880

TELE: ConnectionId=[0x19BDF910 0xAF500007 0x0 0x58ED0] TxDuration=16640
VoiceTxDuration=16640 FaxTxDuration=0 CoderTypeRate=0 NoiseLevel=0 ACOMLevel=4
OutSignalLevel=-440 InSignalLevel=-440 InfoActivity=2 ERLLevel=227
SessionTarget=
```

Table 4-3 provides an alphabetical listing of the fields in this output and a description of each field.

Table 4-3 Show Call Active Voice Field Descriptions

Field	Description
ACOM Level	Current ACOM level for the call. This value is sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
CallOrigin	Call origin; answer versus originate.
CallState	Current state of the call.
CoderTypeRate	Negotiated coder transmit rate of voice/fax compression during the call.
ConnectionId	Global call identifier of a gateway call.
ConnectTime	Time at which the call was connected.
Dial-Peer	Tag of the dial peer transmitting this call.
ERLLevel	Current Echo Return Loss (ERL) level for this call.
FaxTxDuration	Duration of fax transmission from this peer to voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWith Silence	Duration of voice signal replaced with silence because voice data was lost or not received on time for this call.
GapFillWithPrediction	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding in time because voice data was lost or not received in time from the voice gateway for this call. An example of such pullout is frame-eraser or frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithInterpolation	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because voice data was lost or not received on time from voice gateway for this call.

Table 4-3 Show Call Active Voice Field Descriptions (continued)

Field	Description
GapFillWith Redundancy	Duration of voice signal played out with signal synthesized from redundancy parameters available because voice data was lost or not received on time from voice gateway for this call.
HiWaterPlayoutDelay	High water mark Voice Playout FIFO Delay during this call.
Index	Dial-peer identification number.
InfoActivity	Active information transfer activity state for this call.
InfoType	Information type for this call.
InSignalLevel	Active input signal level from the telephony interface used by this call.
LogicalIfIndex	Index number of the logical interface for this call.
LoWaterPlayoutDelay	Low water mark Voice Playout FIFO Delay during the call.
NoiseLevel	Active noise level for the call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
OutSignalLevel	Active output signal level to telephony interface used by this call.
PeerAddress	Destination pattern associated with this peer.
PeerId	ID value of the peer table entry to which this call was made.
PeerIfIndex	Voice-port index number for this peer.
PeerSubaddress	Subaddress to which this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the decoder delay during the voice call.
ReceivePackets	Number of packets received by this peer during this call.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system UDP listener port to which voice packets are transmitted.
RoundTripDelay	Voice packet round trip delay between the local and remote system on the IP backbone during the call.
SelectedQoS	Selected RSVP quality of service (QoS) for the call.
SessionProtocol	Session protocol used for an Internet call between the local and remote router via the IP backbone.
SessionTarget	Session target of the peer used for the call.
SetupTime	Value of the System UpTime when the call associated with this entry was started.
TransmitBytes	Number of bytes transmitted from this peer during the call.
TransmitPackets	Number of packets transmitted from this peer during the call.
TxDuration	Duration of transmit path open from this peer to the voice gateway for the call.
VADEnable	Whether or not voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmission from this peer to voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration value by the TxDuration value.

Related Commands

You can use the master index or search online to find documentation of related commands.

show call history voice
show dial-peer voice
show num-exp
show voice port

show call history voice

To display the call history table, use the **show call history voice** privileged EXEC command.

show call history voice last *number*

Syntax Description

last *number*	Displays the last calls connected, where the number of calls displayed is defined by the argument *number.* A valid entry for the argument *number* is any number from 1 to 2147483647.

Command Mode

Privileged EXEC

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **show call history voice** privileged EXEC command to display the call history table. The call history table contains a listing of all calls connected through this router in descending time order since Voice over IP was enabled. You can display subsets of the call history table by using specific keywords. To display the last calls connected through this router, use the keyword **last**, and define the number of calls to be displayed with the argument *number.*

Sample Display

The following is sample output from the **show call history voice** command:

```
sloth# show call history voice
GENERIC: SetupTime=20405 Index=0 PeerAddress= PeerSubAddress= PeerId=0
PeerIfIndex=0 LogicalIfIndex=0 DisconnectCause=NORMAL DisconnectText= ConnectTime=0
DisconectTime=20595 CallOrigin=2 ChargedUnits=0 InfoType=0 TransmitPackets=0
TransmitBytes=0 ReceivePackets=0 ReceiveBytes=0

VOIP: ConnectionId[0x19BDF910 0xAF500006 0x0 0x56590] RemoteIPAddress=17635075
RemoteUDPPort=16392 RoundTripDelay=0 SelectedQoS=0 SessionProtocol=1
SessionTarget= OnTimeRvPlayout=0 GapFillWithSilence=0 GapFillWithPrediction=0
GapFillWithInterpolation=0 GapFillWithRedundancy=0 HiWaterPlayoutDelay=0
LoWaterPlayoutDelay=0 ReceiveDelay=0 VADEnable=0 CoderTypeRate=0

TELE: ConnectionId=[0x19BDF910 0xAF500006 0x0 0x56590] TxDuration=3030
VoiceTxDuration=2700 FaxTxDuration=0 CoderTypeRate=0 NoiseLevel=0 ACOMLevel=0
SessionTarget=
```

Table 4-4 provides an alphabetical listing of the fields in this output and a description of each field.

Table 4-4 Show Call History Voice Field Descriptions

Field	Description
ACOMLevel	Average ACOM level for this call. This value is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
CallOrigin	Call origin; answer versus originate.
CoderTypeRate	Negotiated coder rate. This value specifies the transmit rate of voice/fax compression to its associated call leg for the call.
ConnectionID	Global call identifier for the gateway call.
ConnectTime	Time the call was connected.
DisconnectCause	Description explaining why the call was disconnected.
DisconnectText	Descriptive text explaining the disconnect reason.
DisconnectTime	Time the call was disconnected.
FaxDuration	Duration of fax transmitted from this peer to the voice gateway for this call. You can derive the Fax Utilization Rate by dividing this value by the TxDuration value.
GapFillWithSilence	Duration of voice signal replaced with silence because the voice data was lost or not received on time for this call.
GapFillWithPrediction	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because the voice data was lost or not received on time from the voice gateway for this call.
GapFillWithInterpolation	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because the voice data was lost or not received on time from the voice gateway for this call.
GapFillWithRedundancy	Duration of voice signal played out with signal synthesized from redundancy parameters available because the voice data was lost or not received on time from the voice gateway for this call.
HiWaterPlayoutDelay	High water mark Voice Playout FIFO Delay during the voice call.
Index	Index number identifying the voice-peer for this call.
InfoType	Information type for this call.
LogicalIfIndex	Index of the logical voice port for this call.
LoWaterPlayoutDelay	Low water mark Voice Playout FIFO Delay during the voice call.
NoiseLevel	Average noise level for this call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
PeerAddress	Destination pattern or number to which this call is connected.
PeerId	ID value of the peer entry table to which this call was made.
PeerIfIndex	Index number of the logical interface through which this call was made. For ISDN media, this would be the index number of the B channel used for the call.
PeerSubAddress	Subaddress to which this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the decoder delay during the voice call.
ReceivePackets	Number of packets received by this peer during the call.

Table 4-4 Show Call History Voice Field Descriptions (continued)

Field	Description
RemoteIPAddress	Remote system IP address for the call.
RemoteUDPPort	Remote system UDP listener port to which voice packets for this call are transmitted.
RoundTripDelay	Voice packet round trip delay between the local and remote system on the IP backbone for this call.
SelectedQoS	Selected RSVP quality of service for the call.
Session Protocol	Session protocol to be used for an Internet call between the local and remote router via the IP backbone.
Session Target	Session target of the peer used for the call.
SetUpTime	Value of the System UpTime when the call associated with this entry was started.
TransmitBytes	Number of bytes transmitted by this peer during the call.
TransmitPackets	Number of packets transmitted by this peer during the call.
TxDuration	Duration of the transmit path open from this peer to the voice gateway for the call.
VADEnable	Whether or not voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmitted from this peer to voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration by the TxDuration value.

Related Commands

You can use the master index or search online to find documentation of related commands.

show call active voice
show dial-peer voice
show num-exp
show voice port

show dial-peer voice

To display configuration information for dial peers, use the **show dial-peer voice** privileged EXEC command.

show dial-peer voice [*number*]

Syntax Description

number	Displays configuration for the dial peer identified by the argument *number*. Valid entries are any integers that identify a specific dial peer, from 1 to 32767.

Command Mode

Privileged EXEC

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **show dial-peer voice** privileged EXEC command to display the configuration for all VoIP and POTS dial peers configured for the router. To show configuration information for only one specific dial peer, use the argument *number* to identify the dial peer.

Sample Display

The following is sample output from the **show dial-peer voice** command for a POTS dial peer:

```
sloth# show dial-peer voice 1
VoiceEncapPeer1
        tag = 1, dest-pat = `+14085551000',
        answer-address = `',
        group = 0, Admin state is up, Operation state is down
        Permission is Both,
        type = pots, prefix = `',
        session-target = `', voice-port =
        Connect Time = 0, Charged Units = 0
        Successful Calls = 0, Failed Calls = 0
        Accepted Calls = 0, Refused Calls = 0
        Last Disconnect Cause is “”
        Last Disconnect Text is “”
        Last Setup Time = 0
```

The following is sample output from the **show dial-peer voice** command for a VoIP dial peer:

```
sloth# show dial-peer voice 10
VoiceOverIpPeer10
        tag = 10, dest-pat = `',
        incall-number = `+14087',
        group = 0, Admin state is up, Operation state is down
        Permission is Answer,
        type = voip, session-target = `',
        sess-proto = cisco, req-qos = bestEffort,
        acc-qos = bestEffort,
        fax-rate = voice, codec = g729r8,
        Expect factor = 10,Icpif = 30, VAD = disabled, Poor QOV Trap = disabled,
        Connect Time = 0, Charged Units = 0
        Successful Calls = 0, Failed Calls = 0
        Accepted Calls = 0, Refused Calls = 0
        Last Disconnect Cause is “”
        Last Disconnect Text is “”
        Last Setup Time = 0
```

Table 4-5 explains the fields contained in both of these examples.

Table 4-5 Show Dial-Peer Voice Field Descriptions

Field	Description
AcceptedCalls	Number of calls from this peer accepted since system startup.
acc-qos	Lowest acceptable quality of service configured for calls for this peer.
Admin state	Administrative state of this peer.
Charged Units	Total number of charging units applying to this peer since system startup.The unit of measure is in hundredths of seconds.
codec	Default voice coder rate of speech for this peer.
Connect Time	Accumulated connect time to the peer since system startup for both incoming and outgoing calls. The unit of value is in hundredths of seconds.
dest-pat	Destination pattern (telephone number) for this peer.
Expect factor	User-requested Expectation Factor of voice quality for calls via this peer.
fax-rate	Fax transmission rate configured for this peer.
Failed Calls	Number of failed call attempts to this peer since system startup.
group	Group number associated with this peer.
ICPIF	Configured Calculated Planning Impairment Factor (ICPIF) value for calls sent by a dial peer.
incall-number	Full E.164 telephone number to be used to identify the dial peer.
Last Disconnect Cause	Encoded network cause associated with the last call. This value will be updated whenever a call is started or cleared and depends on the interface type and session protocol being used on this interface.
Last Disconnect Text	ASCII text describing the reason for the last call termination.
Last Setup Time	Value of the System Up Time when the last call to this peer was started.
Operation state	Operational state of this peer.
Permission	Configured permission level for this peer.
Poor QOV Trap	Whether Poor Quality of Voice trap messages have been enabled or disabled.
Refused Calls	Number of calls from this peer refused since system startup.

Table 4-5 Show Dial-Peer Voice Field Descriptions (continued)

Field	Description
req-qos	Configured requested quality of service for calls for this dial peer.
session-target	Session target of this peer.
sess-proto	Session protocol to be used for Internet calls between local and remote router via the IP backbone.
Successful Calls	Number of completed calls to this peer.
tag	Unique dial-peer ID number.
VAD	Whether or not voice activation detection (VAD) is enabled for this dial peer.

Related Commands

You can use the master index or search online to find documentation of related commands.

show call active voice
show call-history voice
show num-exp
show voice port

show dialplan incall number

To pair different voice ports and telephone numbers together for troubleshooting, use the **show dialplan incall number** privileged EXEC command.

show dialplan incall *slot-number/subunit-number/port* **number** *dial string*

Syntax Description

slot-number/	Specifies the slot number in the Cisco router where the voice network module is installed. Valid entries are from 0 to 3, depending on the voice interface card you have installed.
subunit-number/	Specifies the subunit on the voice network module where the voice port is located. Valid entries are 0 or 1.
port	Specifies the voice port. Valid entries are 0 or 1.
dial string	Specifies a particular destination pattern (telephone number).

Command Mode

Privileged EXEC

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Occasionally, an incoming call cannot be matched to a dial peer in the dial-peer database. One reason this might occur is that the specified destination cannot be reached via the voice interface through which the incoming call came. Use the **show dialplan incall number** command as a troubleshooting method to resolve the call destination by pairing voice ports and telephone numbers together until there is a match.

Example

The following example tests whether the telephone extension 57681 can be reached through voice port 1/0/1:

```
show dialplan incall 1/0/1 number 57681
```

Related Commands

You can use the master index or search online to find documentation of related commands.

show dialplan number

show dialplan number

To show which dial peer is reached when a particular telephone number is dialed, use the **show dial plan number** privileged EXEC command.

show dial plan number *dial string*

Syntax Description

dial string	Specifies a particular destination pattern (telephone number).

Command Mode

Privileged EXEC

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Example

The following example displays the dial peer associated with the destination pattern of 54567:

```
show dialplan number 54567
```

Related Commands

You can use the master index or search online to find documentation of related commands.

show dialplan incall number

show num-exp

To show the number expansions configured, use the **show num-exp** privileged EXEC command.

show num-exp [*dialed- number*]

Syntax Description

dialed-number	Displays number expansion for the specified dialed number.

Command Mode

Privileged EXEC

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **show num-exp** privileged EXEC command to display all of the number expansions configured for this router. To display number expansion for only one number, specify that number by using the *dialed-number* argument.

Sample Display

The following is sample output from the **show num-exp** command:

```
sloth# show num-exp
Dest Digit Pattern = '0...'     Translation = '+14085550...'
Dest Digit Pattern = '1...'     Translation = '+14085551...'
Dest Digit Pattern = '3..'      Translation = '+140855503..'
Dest Digit Pattern = '4..'      Translation = '+140855504..'
Dest Digit Pattern = '5..'      Translation = '+140855505..'
Dest Digit Pattern = '6....'    Translation = '+1408555....'
Dest Digit Pattern = '7....'    Translation = '+1408555....'
Dest Digit Pattern = '8...'     Translation = '+14085558...'
```

Table 4-6 explains the fields in the sample output.

Table 4-6 Show Dial-Peer Voice Field Descriptions

Field	Description
Dest Digit Pattern	Index number identifying the destination telephone number digit pattern.
Translation	Expanded destination telephone number digit pattern.

Related Commands

You can use the master index or search online to find documentation of related commands.

show call active voice
show call history voice
show dial-peer voice
show voice port

show voice port

To display configuration information about a specific voice port, use the **show voice port** privileged EXEC command.

show voice port *slot-number*/*subunit-number*/*port*

Syntax Description

slot-number/	Specifies the slot number in the Cisco router where the voice interface card is installed. Valid entries are from 0 to 3, depending on the slot where it has been installed.
subunit-number/	Specifies the subunit on the voice interface card where the voice port is located. Valid entries are 0 or 1.
port	Specifies the voice port. Valid entries are 0 or 1.

Command Mode

Privileged EXEC

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **show voice port** privileged EXEC command to display configuration and voice interface card-specific information about a specific port.

Sample Display

The following is sample output from the **show voice port** command for an E&M voice port:

```
sloth# show voice port 1/0/0
E&M Slot is 1, Sub-unit is 0, Port is 0
 Type of VoicePort is E&M
 Operation State is unknown
 Administrative State is unknown
 The Interface Down Failure Cause is 0
 Alias is NULL
 Noise Regeneration is disabled
 Non Linear Processing is disabled
 Music On Hold Threshold is Set to 0 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is disabled
 Echo Cancel Coverage is set to 16ms
 Connection Mode is Normal
 Connection Number is
 Initial Time Out is set to 0 s
 Interdigit Time Out is set to 0 s
 Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
```

```
 Voice card specific Info Follows:
 Signal Type is wink-start
 Operation Type is 2-wire
 Impedance is set to 600r Ohm
 E&M Type is unknown
 Dial Type is dtmf
 In Seizure is inactive
 Out Seizure is inactive
 Digit Duration Timing is set to 0 ms
 InterDigit Duration Timing is set to 0 ms
 Pulse Rate Timing is set to 0 pulses/second
 InterDigit Pulse Duration Timing is set to 0 ms
 Clear Wait Duration Timing is set to 0 ms
 Wink Wait Duration Timing is set to 0 ms
 Wink Duration Timing is set to 0 ms
 Delay Start Timing is set to 0 ms
 Delay Duration Timing is set to 0 ms
```

The following is sample output from the **show voice port** command for an FXS voice port:

```
sloth# show voice port 1/0/0
Foreign Exchange Station 1/0/0 Slot is 1, Sub-unit is 0, Port is 0
 Type of VoicePort is FXS
 Operation State is DORMANT
 Administrative State is UP
 The Interface Down Failure Cause is 0
 Alias is NULL
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to 0 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 16ms
 Connection Mode is Normal
 Connection Number is
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
 Voice card specific Info Follows:
 Signal Type is loopStart
 Ring Frequency is 25 Hz
 Hook Status is On Hook
 Ring Active Status is inactive
 Ring Ground Status is inactive
 Tip Ground Status is inactive
 Digit Duration Timing is set to 100 ms
 InterDigit Duration Timing is set to 100 ms
 Hook Flash Duration Timing is set to 600 ms
```

Table 4-7 explains the fields in the sample output.

Table 4-7 Show Voice Port Field Descriptions

Field	Description
Administrative State	Administrative state of the voice port.
Alias	User-supplied alias for this voice port.
Clear Wait Duration Timing	Time of inactive seizure signal to declare call cleared.
Connection Mode	Connection mode of the interface.
Connection Number	Full E.164 telephone number used to establish a connection with the trunk or PLAR mode.
Currently Processing	Type of call currently being processed: none, voice, or fax.
Delay Duration Timing	Maximum delay signal duration for delay dial signaling.
Delay Start Timing	Timing of generation of delayed start signal from detection of incoming seizure.
Dial Type	Out-dialing type of the voice port.
Digit Duration Timing	DTMF Digit duration in milliseconds.
E&M Type	Type of E&M interface.
Echo Cancel Coverage	Echo Cancel Coverage for this port.
Echo Cancellation	Whether or not echo cancellation is enabled for this port.
Hook Flash Duration Timing	Maximum length of hook flash signal.
Hook Status	Hook status of the FXO/FXS interface.
Impedance	Configured terminating impedance for the E&M interface.
In Gain	Amount of gain inserted at the receiver side of the interface.
In Seizure	Incoming seizure state of the E&M interface.
Initial Time Out	Amount of time the system waits for an initial input digit from the caller.
InterDigit Duration Timing	DTMF interdigit duration in milliseconds.
InterDigit Pulse Duration Timing	Pulse dialing interdigit timing in milliseconds.
Interdigit Time Out	Amount of time the system waits for a subsequent input digit from the caller.
Maintenance Mode	Maintenance mode of the voice-port.
Music On Hold Threshold	Configured Music-On-Hold Threshold value for this interface.
Noise Regeneration	Whether or not background noise should be played to fill silent gaps if VAD is activated.
Number of signaling protocol errors	Number of signaling protocol errors.
Non-Linear Processing	Whether or not Non-Linear Processing is enabled for this port.
Operations State	Operation state of the port.
Operation Type	Operation of the E&M signal: 2-wire or 4-wire.
Out Attenuation	Amount of attenuation inserted at the transmit side of the interface.
Out Seizure	Outgoing seizure state of the E&M interface.
Port	Port number for this interface associated with the voice interface card.
Pulse Rate Timing	Pulse dialing rate in pulses per second (pps).
Regional Tone	Configured regional tone for this interface.

Table 4-7 Show Voice Port Field Descriptions (continued)

Field	Description
Ring Active Status	Ring active indication.
Ring Frequency	Configured ring frequency for this interface.
Ring Ground Status	Ring ground indication.
Signal Type	Type of signaling for a voice port: loop-start, ground-start, wink-start, immediate, and delay-dial.
Slot	Slot used in the voice interface card for this port.
Sub-unit	Subunit used in the voice interface card for this port.
Tip Ground Status	Tip ground indication.
Type of VoicePort	Type of voice port: FXO, FXS, and E&M.
The Interface Down Failure Cause	Text string describing why the interface is down,
Wink Duration Timing	Maximum wink duration for wink start signaling.
Wink Wait Duration Timing	Maximum wink wait duration for wink start signaling.

Related Commands

You can use the master index or search online to find documentation of related commands.

show call active voice
show call history voice
show dial-peer voice
show num-exp

shutdown (dial-peer configuration)

To change the administrative state of the selected dial peer from up to down, use the **shutdown** dial-peer configuration command. Use the **no** form of this command to change the administrative state of this dial peer from down to up.

shutdown
no shutdown

Syntax Description

This command has no arguments or keywords.

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

When a dial peer is shut down, you cannot initiate calls to that peer. This command is applicable to both VoIP and POTS peers.

Example

The following example changes the administrative state of voice telephony dial peer 10 to down:

```
configure terminal
 dial-peer voice 10 pots
 shutdown
```

shutdown (voice-port configuration)

To take the voice ports for a specific voice interface card offline, use the **shutdown** voice-port configuration command. Use the **no** form of this command to put the ports back in service.

shutdown
no shutdown

Syntax Description

This command has no arguments or keywords.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

When you enter the **shutdown** command, all ports on the voice interface card are disabled. When you enter the **no shutdown** command, all ports on the voice interface card are enabled. A telephone connected to an interface will hear dead silence when a port is shut down.

Example

The following example takes voice port 1/1/0 offline:

```
configure terminal
 voice port 1/1/0
 shutdown
```

Note The preceding configuration example will shut down both voice ports 1/1/0 and 1/1/1.

signal

To specify the type of signaling for a voice port, use the **signal** voice-port configuration command. Use the **no** form of this command to restore the default value for this command.

signal {**loop-start** | **ground-start** | **wink-start** | **immediate** | **delay-dial**}
no signal

Syntax Description

loop-start	Specifies Loop Start signaling. Used for FXO and FXS interfaces. With Loop Start signaling only one side of a connection can hang up. This is the default setting for FXO and FXS voice ports.
ground-start	Specifies Ground Start signaling. Used for FXO and FXS interfaces. Ground Start allows both sides of a connection to place a call and to hang up.
wink-start	Indicates that the calling side seizes the line by going off-hook on its E lead then waits for a short off-hook "wink" indication on its M lead from the called side before sending address information as DTMF digits. Used for E&M tie trunk interfaces. This is the default setting for E&M voice ports.
immediate	Indicates that the calling side seizes the line by going off-hook on its E lead and sends address information as DTMF digits. Used for E&M tie trunk interfaces.
delay-dial	Indicates that the calling side seizes the line by going off-hook on its E lead. After a timing interval, the calling side looks at the supervision from the called side. If the supervision is on-hook, the calling side starts sending information as DTMF digits; otherwise, the calling side waits until the called side goes on-hook and then starts sending address information. Used for E&M tie trunk interfaces.

Default

The default value is **loop-start** for FXO and FXS interfaces. The default value is **wink-start** for E&M interfaces.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Configuring the **signal** command for an FXS or FXO voice port will change the signal value for both voice ports on a VPM card.

Note If you change the signal type for an FXO voice port, you need to move the appropriate jumper in the voice interface card of the voice network module. For more information about the physical characteristics of the voice network module, refer to the *Voice Network Module and Voice Interface Card Configuration Note* that came with your voice network module

---

Configuring this command for an E&M voice port will change only the signal value for the selected voice port. In either case, the voice port must be shut down and then activated before the configured values will take effect.

Some PBXes will miss initial digits if the E&M voice port is configured for Immediate signaling. If this occurs, use Delay-Dial signaling instead. Some non-Cisco devices have a limited number of DTMF receivers. This type of equipment must delay the calling side until a DTMF receiver is available.

Example

The following example configures Ground Start signaling, which means that both sides of a connection can place a call and hang up, as the signaling type for a voice port:

```
configure terminal
 voice-port 1/1/1
 signal ground-start
```

snmp enable peer-trap poor-qov

To generate poor quality of voice notification for applicable calls associated with VoIP dial peers, use the **snmp enable peer-trap poor-qov** dial-peer configuration command. Use the **no** form of this command to disable this feature.

snmp enable peer-trap poor-qov
no snmp enable peer-trap poor-qov

Syntax Description

This command has no arguments or keywords.

Default

Disabled

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **snmp enable peer-trap poor qov** command to generate poor quality of voice notifications for applicable calls associated with this dial peer. If you have an SNMP manager that will use SNMP messages when voice quality drops, you might want to enable this command. Otherwise, you should disable this command to reduce unnecessary network traffic.

This command is applicable only to VoIP peers.

Example

The following example enables poor quality of voice notifications for calls associated with VoIP dial peer 10:

```
dial-peer voice 10 voip
 snmp enable peer-trap poor-qov
```

Related Commands

You can use the master index or search online to find documentation of related commands.

snmp-server enable trap voice poor-qov
snmp trap link-status

snmp-server enable traps

To enable the router to send SNMP traps, use the **snmp-server enable traps** global configuration command. Use the **no** form of this command to disable SNMP traps.

snmp-server enable traps [*trap-type*] [*trap-option*]
no snmp-server enable traps [*trap-type*] [*trap-option*]

Syntax Description

trap-type	(Optional) Type of trap to enable. If no type is specified, all traps are sent (including the **envmon** and **repeater** traps). The trap type can be one of the following keywords: • **bgp**—Sends Border Gateway Protocol (BGP) state change traps. • **config**—Sends configuration traps. • **entity**—Sends Entity MIB modification traps. • **envmon**—Sends Cisco enterprise-specific environmental monitor traps when an environmental threshold is exceeded. When the **envmon** keyword is used, you can specify a *trap-option* value. • **frame-relay**—Sends Frame Relay traps. • **isdn**—Sends Integrated Services Digital Network (ISDN) traps. When the **isdn** keyword is used on Cisco 1600 series routers, you can specify a *trap-option* value. • **repeater**—Sends Ethernet hub repeater traps. When the **repeater** keyword is selected, you can specify a *trap-option* value. • **rtr**—Sends response time reporter (RTR) traps. • **snmp**—Sends Simple Network Management Protocol (SNMP) traps. When the **snmp** keyword is used, you can specify a *trap-option* value. • **syslog**—Sends error message traps (Cisco Syslog MIB). Specify the level of messages to be sent with the **logging history level** command. • **voice**—Sends SNMP poor quality of voice traps, when used with the **qov** *trap-option*.

trap-option	(Optional) When the **envmon** keyword is used, you can enable a specific environmental trap type, or accept all trap types from the environmental monitor system. If no option is specified, all environmental types are enabled. The option can be one or more of the following keywords: **voltage**, **shutdown**, **supply**, **fan**, and **temperature**. When the **isdn** keyword is used on Cisco 1600 series routers, you can specify the **call-information** keyword to enable an SNMP ISDN call information trap for the ISDN MIB subsystem, or you can specify the **isdnu-interface** keyword to enable an SNMP ISDN U interface trap for the ISDN U interface MIB subsystem. When the **repeater** keyword is used, you can specify the repeater option. If no option is specified, all repeater types are enabled. The option can be one or more of the following keywords: • **health**—Enables IETF Repeater Hub MIB (RFC 1516) health trap. • **reset**—Enables IETF Repeater Hub MIB (RFC 1516) reset trap. When the **snmp** keyword is used, you can specify the **authentication** option to enable SNMP Authentication Failure traps. (The **snmp-server enable traps snmp authentication** command replaces the **snmp-server trap-authentication** command.) If no option is specified, all SNMP traps are enabled. When the **voice** keyword is used, you can enable SNMP poor quality of voice traps by using the **qov** option.

Defaults

This command is disabled by default. No traps are enabled.

Some trap types cannot be controlled with this command. These traps are either always enabled or enabled by some other means. For example, the linkUpDown messages are disabled by the **no snmp trap link-status** command.

If you enter this command with no keywords, the default is to enable all trap types.

Command Mode

Global configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.1.

This command is useful for disabling traps that are generating a large amount of uninteresting or useless noise.

If you do not enter an **snmp-server enable traps** command, no traps controlled by this command are sent. In order to configure the router to send these SNMP traps, you must enter at least one **snmp-server enable traps** command. If you enter the command with no keywords, all trap types are enabled. If you enter the command with a keyword, only the trap type related to that keyword is enabled. In order to enable multiple types of traps, you must enter a separate **snmp-server enable traps** command for each trap type and option.

The **snmp-server enable traps** command is used in conjunction with the **snmp-server host** command. Use the **snmp-server host** command to specify which host or hosts receive SNMP traps. In order to send traps, you must configure at least one **snmp-server host** command.

For a host to receive a trap controlled by this command, both the **snmp-server enable traps** command and the **snmp-server host** command for that host must be enabled. If the trap type is not controlled by this command, just the appropriate **snmp-server host** command must be enabled.

The trap types used in this command all have an associated MIB object that allows them to be globally enabled or disabled. Not all of the trap types available in the **snmp-server host** command have notificationEnable MIB objects, so some of these cannot be controlled using the **snmp-server enable traps** command.

Examples

The following example enables the router to send SNMP poor quality of voice traps:

```
configure terminal
 snmp-server enable trap voice poor-qov
```

The following example enables the router to send all traps to the host myhost.cisco.com using the community string *public*:

```
snmp-server enable traps
snmp-server host myhost.cisco.com public
```

The following example enables the router to send Frame Relay and environmental monitor traps to the host myhost.cisco.com using the community string *public*:

```
snmp-server enable traps frame-relay
snmp-server enable traps envmon temperature
snmp-server host myhost.cisco.com public
```

The following example will not send traps to any host. The BGP traps are enabled for all hosts, but the only traps enabled to be sent to a host are ISDN traps.

```
snmp-server enable traps bgp
snmp-server host bob public isdn
```

Related Commands

You can use the master indexes or search online to find documentation of related commands.

snmp enable peer-trap peer-qov
snmp-server host
snmp-server trap-source
snmp trap illegal-address
snmp trap link-status

snmp trap link-status

To enable Simple Network Management Protocol (SNMP) trap messages to be generated when this voice port is brought up or down, use the **snmp trap link-status** voice-port configuration command. Use the **no** form of this command to disable this feature.

snmp trap link-status
no snmp trap link-status

Syntax Description

This command contains no arguments or keywords.

Default

The default for this command is enabled.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **snmp trap link-status** command to enable SNMP trap messages (linkup and linkdown) to be generated whenever this voice port is brought online or offline.

If you are managing the equipment with an SNMP manager, this command should be enabled. Enabling link-status messages will allow the SNMP manager to learn of a status change without polling the equipment. If you are not using an SNMP manager, this command should be disabled to avoid unnecessary network traffic.

Example

The following example enables SNMP trap messages for voice-port 2/1/0:

```
voice-port 2/1/0
 snmp trap link-stat
```

Related Commands

You can use the master index or search online to find documentation of related commands.

smnp enable peer-trap poor-qov
snmp-server enable trap poor-qov

timeouts initial

To configure the initial digit timeout value for a specified voice port, use the **timeouts initial** voice-port configuration command. Use the **no** form of this command to restore the default value for this command.

timeouts initial *seconds*
no timeouts initial *seconds*

Syntax Description

initial *seconds*	Specifies the initial timeout duration in seconds. Valid entries are any integer from 0 to 120.

Default

The default value is 10 seconds.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **timeouts initial** command to specify the number of seconds the system will wait for the caller to input the first digit of the dialed digits. The timeouts initial timer is activated when the call is accepted and is deactivated when the caller inputs the first digit. If the configured timeout value is exceeded, the caller is notified through the appropriate tone and the call is terminated.

To disable the timeouts initial timer, set the *seconds* value to **0**.

Example

The following example sets the initial digit timeout value to 15 seconds:

```
voice-port 1/0/0
 timeouts initial 15
```

Related Commands

You can use the master index or search online to find documentation of related commands.

timeouts interdigit
timing

timeouts interdigit

To configure the interdigit timeout value for a specified voice port, use the **timeouts interdigit** voice-port configuration command. Use the **no** form of this command to restore the default value for this command.

timeouts interdigit *seconds*
no timeouts interdigit *seconds*

Syntax Description

seconds	Specifies the interdigit timeout duration in seconds. Valid entries are any integer from 0 to 120.

Default

The default value is 10 seconds.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **timeouts interdigit** command to specify the number of seconds the system will wait (after the caller has input the initial digit) for the caller to input a subsequent digit of the dialed digits. The timeouts interdigit timer is activated when the caller inputs a digit and restarted each time the caller inputs another digit until the destination address is identified. If the configured timeout value is exceeded before the destination address is identified, the caller is notified through the appropriate tone and the call is terminated.

To disable the timeouts interdigit timer, set the *seconds* value to **0**.

Example

The following example sets the interdigit timeout value for 15 seconds:

```
voice-port 1/0/0
 timeouts interdigit 15
```

Related Commands

You can use the master index or search online to find documentation of related commands.

timeouts initial
timing

timing clear-wait

To indicate the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port, use the **timing clear-wait** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing clear-wait *milliseconds*
no timing clear-wait *milliseconds*

Syntax Description

milliseconds	Minimum amount of time, in milliseconds, between the inactive seizure signal and the call being cleared. Valid entries on the Cisco 3600 series are numbers from 200 to 2000. Supported on E&M ports only.

Default

400 milliseconds

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Examples

The following example configures the clear-wait duration on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing clear-wait 300
```

Related Commands

You can use the master indexes or search online to find documentation of related commands.

timeouts initial
timeouts interdigit
timing clear-wait
timing delay-duration
timing delay-start
timing dial-pulse min-delay
timing digit
timing interdigit
timing pulse
timing pulse-interdigit
timing wink-duration
timing wink-wait

timing delay-duration

To specify the delay signal duration for a specified voice port, use the **timing delay-duration** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing delay-duration *milliseconds*
no timing delay-duration *milliseconds*

Syntax Description

milliseconds	Delay signal duration for delay dial signaling, in milliseconds. Valid entries are numbers from 100 to 5000. Supported on E&M ports only.

Default

2000 milliseconds

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call direction for the **timing delay-duration** command is out.

Examples

The following example configures the delay signal duration on a Cisco 3600 series voice port to 3000 milliseconds:

```
voice-port 1/0/0
 timing delay-duration 3000
```

Related Commands

You can use the master indexes or search online to find documentation of related commands.

timeout initial
timeouts interdigit
timing clear-wait
timing delay-start
timing dial-pulse min-delay
timing digit
timing interdigit
timing pulse
timing pulse-interdigit
timing wink-duration
timing wink-wait

timing delay-start

To specify the minimum delay time from outgoing seizure to out-dial address for a specified voice port, use the **timing delay-start** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing delay-start *milliseconds*
no timing delay-start *milliseconds*

Syntax Description

milliseconds	Minimum delay time, in milliseconds, from outgoing seizure to out-dial address. Valid entries are numbers from 20 to 2000. Supported on E&M ports only.

Default

300 milliseconds

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call direction for the **timing delay-start** command is out.

Examples

The following example configures the delay-start duration on a Cisco 3600 series voice port to 250 milliseconds:

```
voice-port 1/0/0
 timing delay-start 250
```

Related Commands

You can use the master indexes or search online to find documentation of related commands.

timeouts initial
timeouts interdigit
timing clear-wait
timing delay-duration
timing dial-pulse min-delay
timing digit
timing interdigit
timing pulse
timing pulse-interdigit
timing wink-duration
timing wink-wait

timing dial-pulse min-delay

To specify the time between wink-like pulses for a specified voice port on the Cisco 3600 series, use the **timing dial-pulse min-delay** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing dial-pulse min-delay *milliseconds*
no timing dial-pulse min-delay *milliseconds*

Syntax Description

milliseconds	Time, in milliseconds, between the generation of wink-like pulses. Valid entries are numbers from 0 to 5000.

Default

300 milliseconds

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **timing** command with the **dial-pulse min-delay** keyword with PBXes requiring a wink-like pulse, even though they have been configured for delay-dial signaling. If the value for this keyword is set to 0, the router will not generate this wink-like pulse. The call signal direction for this command is in.

Example

The following example configures the time between the generation of wink-like pulses on a Cisco 3600 series voice port to 350 milliseconds:

```
voice-port 1/0/0
 timing dial-pulse min-delay 350
```

Related Commands

You can use the master indexes or search online to find documentation of related commands.

timeouts initial
timeouts interdigit
timing clear-wait
timing delay-duration
timing dialout-delay
timing digit
timing interdigit
timing pulse
timing pulse-interdigit
timing wink-duration
timing wink-wait

timing digit

To specify the DTMF digit signal duration for a specified voice port, use the **timing digit** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing digit *milliseconds*
no timing digit *milliseconds*

Syntax Description

milliseconds	The DTMF digit signal duration, in milliseconds. Valid entries are numbers from 50 to 100. Supported on FXO, FXS and E&M ports.

Default

100 milliseconds

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call signal direction for the **timing digit** command is out.

Examples

The following example configures the DTMF digit signal duration on a Cisco 3600 series voice port to 50 milliseconds:

```
voice-port 1/0/0
 timing digit 50
```

Related Commands

You can use the master indexes or search online to find documentation of related commands.

timeouts initial
timeouts interdigit
timing clear-wait
timing delay-duration
timing delay-start
timing dial-pulse min-delay
timing interdigit
timing pulse
timing pulse-interdigit
timing wink-duration
timing wink-wait

timing interdigit

To specify the DTMF interdigit duration for a specified voice port, use the **timing interdigit** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing interdigit *milliseconds*
no timing interdigit *milliseconds*

Syntax Description

milliseconds	DTMF interdigit duration, in milliseconds. Valid entries are numbers from 50 to 500. Supported on FXO, FXS and E&M ports.

Default

100 milliseconds

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call signal direction for the **timing interdigit** command is out.

Examples

The following example configures the DTMF interdigit duration on a Cisco 3600 series voice port to 150 milliseconds:

```
voice-port 1/0/0
 timing interdigit 150
```

Related Commands

You can use the master indexes or search online to find documentation of related commands.

timeouts initial
timeouts interdigit
timing clear-wait
timing delay-duration
timing delay-start
timing dial-pulse min-delay
timing digit
timing pulse
timing pulse-interdigit
timing wink-duration
timing wink-wait

timing pulse

To specify the pulse dialing rate for a specified voice port, use the **timing pulse** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing pulse *pulses-per-second*
no timing pulse *pulses-per-second*

Syntax Description

pulses-per-second	Pulse dialing rate, in pulses per second. Valid entries are numbers from 10 to 20. Supported on FXO and E&M ports only.

Default

20

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call signal direction for the **timing pulse** command is out.

Examples

The following example configures the pulse dialing rate on a Cisco 3600 series voice port to 15 pulses per second:

```
voice-port 1/0/0
 timing pulse 15
```

Related Commands

You can use the master indexes or search online to find documentation of related commands.

timeouts initial
timeouts interdigit
timing clear-wait
timing delay-duration
timing delay-start
timing dial-pulse min-delay
timing digit
timing interdigit
timing pulse-interdigit
timing wink-duration
timing wink-wait

timing pulse-interdigit

To specify the pulse interdigit timing for a specified voice port, use the **timing pulse-interdigit** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing pulse-interdigit *milliseconds*
no timing pulse-interdigit *milliseconds*

Syntax Description

milliseconds	Pulse dialing interdigit timing, in milliseconds. Valid entries are numbers from 100 to 1000. Supported on FXO and E&M ports only.

Default

500 milliseconds

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call signal direction for the **timing pulse-interdigit** command is out.

Examples

The following example configures the pulse-dialing interdigit timing on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing pulse-interdigit 300
```

Related Commands

You can use the master indexes or search online to find documentation of related commands.

timeout initial
timeouts interdigit
timing clear-wait
timing delay-duration
timing delay-start
timing dial-pulse min-delay
timing digit
timing interdigit
timing pulse
timing wink-duration
timing wink-wait

timing wink-duration

To specify the maximum wink signal duration for a specified voice port, use the **timing wink-duration** voice-port configuration command. Use the **no** form of this command to restore the default value.

timing wink-duration *milliseconds*
no timing wink-duration *milliseconds*

Syntax Description

milliseconds	Maximum wink signal duration, in milliseconds, for a wink-start signal. Valid entries are numbers from 100 to 400. Supported on E&M ports only.

Default

200 milliseconds

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call signal direction for the **timing wink-duration** command is out.

Examples

The following example configures the wink signal duration on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing wink-duration 300
```

Related Commands

You can use the master indexes or search online to find documentation of related commands.

timeouts initial
timeouts interdigit
timing clear-wait
timing delay-duration
timing delay-start
timing dial-pulse min-delay
timing digit
timing interdigit
timing pulse
timing pulse-interdigit
timing wink-wait

timing wink-wait

To specify the maximum wink-wait duration for a specified voice port, use the **timing wink-wait** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing wink-wait *milliseconds*
no timing wink-wait *milliseconds*

Syntax Description

milliseconds	Maximum wink-wait duration, in milliseconds, for a wink start signal. Valid entries are numbers from 100 to 5000. Supported on E&M ports only

Default

200 milliseconds

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call signal direction for the **timing wink-wait** command is out.

Examples

The following example configures the wink-wait duration on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing wink-wait 300
```

Related Commands

You can use the master indexes or search online to find documentation of related commands.

timeouts initial
timeouts interdigit
timing clear-wait
timing delay-duration
timing delay-start
timing dial-pulse min-delay
timing digit
timing interdigit
timing pulse
timing pulse-interdigit
timing wink-duration

type

To specify the E&M interface type, use the **type** voice-port configuration command. Use the **no** form of this command to reset the default value for this command.

type {**1** | **2** | **3** | **5**}
no type

Syntax Description

1	Indicates the following lead configuration: E—output, relay to ground. M—input, referenced to ground.
2	Indicates the following lead configuration: E—output, relay to SG. M—input, referenced to ground. SB—feed for M, connected to -48V. SG—return for E, galvanically isolated from ground.
3	Indicates the following lead configuration: E—output, relay to ground. M—input, referenced to ground. SB—connected to -48V. SG—connected to ground.
5	Indicates the following lead configuration: E—output, relay to ground. M—input, referenced to -48V.

Default

The default value is 1.

Command Mode

Voice-port configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **type** command to specify the E&M interface for a particular voice port. With **1**, the tie-line equipment generates the E-signal to the PBX type grounding the E-lead. The tie line equipment detects the M-signal by detecting current flow to ground. If you select **1**, a common ground must exist between the line equipment and the PBX.

With **2**, the interface requires no common ground between the equipment, thereby avoiding ground loop noise problems. The E-signal is generated toward the PBX by connecting it to SG. M-signal is indicated by the PBX connecting it to SB. While Type 2 interfaces do not require a common ground, they do have the tendency to inject noise into the audio paths because they are asymmetrical with respect to the current flow between devices.

With **3**, the interface operates the same as Type 1 interfaces with respect to the E-signal. The M-signal, however, is indicated by the PBX connecting it to SB on assertion and alternately connecting it to SG during inactivity. If you select **3**, a common ground must be shared between equipment.

With **5**, the Type 5 line equipment indicates E-signal to the PBX by grounding the E-lead. The PBX indicates M-signal by grounding the M-lead. A Type 5 interface is quasi-symmetrical in that while the line is up, current flow is more or less equal between the PBX and the line equipment but noise injection is a problem.

Example

The following example selects Type 3 as the interface type for your voice port:

```
voice-port 1/0/0
 type 3
```

vad

To enable voice activity detection (VAD) for the calls using this dial peer, use the **vad** dial-peer configuration command. Use the **no** form of this command to disable this feature.

vad
no vad

Syntax Description

This command has no arguments or keywords.

Default

Enabled

Command Mode

Dial-peer configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **vad** command to enable voice activity detection. With VAD, silence is not transmitted over the network, only audible speech. If you enable VAD, the sound quality will be slightly degraded but the connection will monopolize much less bandwidth. If you use the **no** form of this command, VAD is disabled and voice data is continuously transmitted to the IP backbone.

This command is applicable only to VoIP peers.

Example

The following example enables VAD:

```
dial-peer voice 10 voip
 vad
```

Related Commands

You can use the master index or search online to find documentation of related commands.

comfort-noise

voice-port

To enter the voice-port configuration mode, use the **voice-port** global configuration command.

voice-port *slot-number/subunit-number/port*

Syntax Description

slot-number/	Specifies the slot number in the Cisco router where the voice network module is installed. Valid entries are from 0 to 3, depending on the voice interface card you have installed.
subunit-number/	Specifies the subunit on the voice network module where the voice port is located. Valid entries are 0 or 1.
port	Specifies the voice port. Valid entries are 0 or 1.

Command Mode

Global configuration

Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **voice-port** global configuration command to switch to the voice-port configuration mode from the global configuration mode. Use the **exit** command to exit the voice-port configuration mode and return to the global configuration mode.

For more information about the physical characteristics of the voice network module, or how to install it, refer to installation documentation that came with your voice network module.

Example

The following example accesses the voice-port configuration mode for port 0, located on subunit 0 on a voice interface card installed in slot 1:

```
configure terminal
 voice-port 1/0/0
```

Related Commands

You can use the master index or search online to find documentation of related commands.

dial-peer